What is claimed is:

1. A method for conditioning a speech signal in preparation for coding of the speech signal, the method comprising the steps of:

accumulating samples of the speech signal over at least a minimum sampling duration;

evaluating the accumulated samples associated with the minimum sampling period to obtain a representative sample;

determining whether a slope of the representative sample of the speech signal conforms to a defined characteristic slope stored in a reference database of spectral characteristics; and

selecting one of a first filter and a second filter for application to the speech signal prior to the coding based on the determination on the slope of the representative sample.

2. The method according to claim 1 where the selecting step comprises selecting the first filter if a slope of the representative sample of the speech signal conforms to the defined characteristic slope in accordance with the determining step.

3. The method according to claim 2 further comprising the step of applying the first filter to lessen a slope of the speech signal to approach a flatter spectral response in preparation for the coding.

4. The method according to claim 1 where the selecting step comprises selecting the second filter if a slope of the representative sample of the speech signal is generally flat in accordance with the determining step.

5. The method according to claim 4 further comprising the step of applying the second filter to increase a slope of the spectral response of the speech signal to

approach a more sloped spectral response than the flat spectral response in preparation for the coding.

6. The method according to claim 1 where the evaluating step comprises averaging the accumulated samples over the minimum sampling duration to obtain the representative sample.

7. The method according to claim 1 further comprising the step of assuming the spectral response of a speech signal is sloped in accordance with the defined characteristic slope prior to completion of at least one of the accumulating step and the determining step.

8. The method according to claim 7 wherein the selecting step comprises selecting the first filter as an initial default filter based on the assumption that the spectral response of the speech signal is sloped in accordance with the defined characteristic slope.

9. The method according to claim 1 where the defined characteristic slope approximately represents a Modified Intermediate Response System.

10. The method according to claim 1 further comprising the step of adjusting at least one encoding parameter to a revised encoding parameter for an encoding process, the at least one encoding parameter affiliated with the selecting of one of the first filter and the second filter.

11. The method according to claim 10 where the adjusting step comprises adjusting an encoding parameter selected from the group consisting of pitch gains per frame or subframe, at least one filter coefficient of a perceptual weighting filter, at least one bandwidth expansion constant associated with a synthesis filter, and at least one bandwidth expansion constant associated with an analysis filter.

12. The method according to claim 1 further comprising the step of adjusting at least one decoding parameter to a revised decoding parameter for a decoding process, the at least one decoding parameter affiliated with the selecting of one of the first filter and the second filter.

13. The method according to claim 12 where the adjusting step comprises adjusting a decoding parameter selected from the group consisting of at least one bandwidth expansion constant associated with a synthesis filter and at least one linear predictive filter coefficient associated with a post filter.

14. The method according to claim 1 further comprising the step of adjusting at least one coding parameter to a revised coding parameter for at least one of an encoding and a decoding process, the at least one coding parameter affiliated with the selecting of one of the first filter and the second filter.

15. The method according to claim 14 where the adjusting step comprises adjusting a coding parameter selected from the group consisting of pitch gains per frame or subframe, at least one filter coefficient of a perceptual weighting filter, at least one bandwidth expansion constant associated with a synthesis filter, at least one bandwidth expansion constant associated with an analysis filter, and at least one linear predictive filter coefficient associated with a post filter.

16. The method according to claim 1 further comprising adjusting a bandwidth expansion of the speech signal to change a value of a linear predictive coefficient for at least one of a synthesis filter and an analysis filter from a previous value to a revised value based on a degree of slope or flatness in the speech signal.

17. The method according to claim 1 further comprising adjusting bandwidth expansion of the speech signal in conformance with the following equations:

$$\frac{1}{A(z)} = \frac{1}{1 - \sum_{i=1}^{P} a_{i\ \mathrm{revised}} z^{-i}},$$

where 1/A(z) is a filter response represented by a z transfer function, ai revised is a linear predictive coefficient, $i = 1 \ldots P$, and P is the prediction order or filter order of the synthesis filter,

$$a_{i\ \mathrm{revised}} = a_{i\ \mathrm{previous}} \gamma^{i},$$

where $a_{i\ \mathrm{revised}}$ is a revised linear predictive coefficient, $a_{i\ \mathrm{previous}}$ is a previous linear predictive coefficient, γ is the bandwidth expansion constant, $i = 1 \ldots P$, and P is the prediction order of the synthesis filter of the encoder, and where $a_{i\ \mathrm{previous}}$ represents a member of the set of extracted linear predictive coefficients $\{a_{i\ \mathrm{previous}}\}^{P}_{i=1}$, for the synthesis filter of the encoder.

18. The method according to claim 17 where the value of the bandwidth expansion constant for a generally flat spectral response differs from that of the defined characteristic slope.

19. The method according to claim 17 where the value of the bandwidth expansion constant is greater for a generally flat spectral response than the defined characteristic slope.

20. The method according to claim 17 where γ is set to a first value of approximately .99 if the slope of the representative sample is consistent with an MIRS spectral response and γ is set to a second value of approximately .995 where the slope of the representative sample is generally flat or approaches zero.

21. The method according to claim 1 further comprising adjusting a frequency response of a perceptual weighting filter based on a degree of slope or flatness in the speech signal.

22. The method according to claim 1 further comprising adjusting a frequency response of a perceptual weighting filter based on the following equation:

$$W(z) = \frac{1}{1 - \alpha z^{-1}} \frac{1 + \sum_{i=1}^{P} a_i \rho^i z^{-i}}{1 + \sum_{i=1}^{P} a_i \beta^i z^{-i}}$$

where α is a weighting constant, β and ρ are preset coefficients, P is the predictive order, and $\{a_i\}$ is the linear predictive coding coefficient.

23. The method according to claim 22 wherein the adjusting step comprises selecting different values of the weighting constant α to adjust the frequency response of the perceptual weighting filter in response to the determined slope or flatness of the speech signal.

24. The method according to claim 22 further comprising controlling the value of α based on the spectral response of the speech signal such that α approximately equals .2 where the speech signal is consistent with the MIRS spectral response and α approximately equals 0 where the speech signal is consistent with a generally flat signal response.

25. The method according to claim 1 further comprising the step of adjusting a frequency response of a post filter coupled to an output of a decoder based on a degree of slope or flatness of the speech signal.

26. The method according to claim 1 further comprising the step of adjusting a frequency response of a post filter in accordance with the following equation:

$$P(z) = \frac{1 + \sum_{i=1}^{P} a_i \gamma_1^i z^{-i}}{1 + \sum_{i=1}^{P} a_i \gamma_2^i z^{-i}}$$

where γ_1 and γ_2 represents a set of post-filtering weighting constants, {ai} is the linear predictive coding coefficient, and P is the filter order of the post filter.

27. The method according to claim 26 further comprising the step of adjusting a frequency response of a post filter by selecting different values of post-filtering weighting constants of γ_1 and γ_2 in response to the determined slope or flatness of the speech signal.

28. The method according to claim 26 where γ_1 and γ_2 approximately equal .65 and .4, respectively, if the speech signal is consistent with an MIRS spectral response; and where γ_1 and γ_2 approximately equal .63 and .4, respectively, if the speech signal is consistent with a generally flat signal response.

29. A system for conditioning a speech signal prior to coding the speech signal, the system comprising:

a buffer memory for accumulating samples of the speech signal over at least a minimum sampling duration;

an averaging unit for evaluating the accumulated samples associated with the minimum sampling period to obtain a representative sample;

a storage device adapted to store spectral characteristics for classifying the speech signal as a closest one of a defined characteristic slope and a flat speech signal;

an evaluator adapted to determine whether a slope of the representative sample of the speech signal conforms to a defined characteristic slope stored in the storage device; and

a selector for selecting a preferential one of a first filter and a second filter for application to the speech signal prior to the coding based on the determination on the slope of the representative sample.

30. The system according to claim 29 where the selector is adapted to select the first filter if the evaluator determines that a slope of the representative sample of the speech signal generally conforms to the defined characteristic slope.

31. The system according to claim 29 where the first filter has a filtering response that lessens a slope of the speech signal to approach a flatter spectral response in preparation for subsequent coding.

32. The system according to claim 29 where the selector is adapted to select the second filter if the evaluator determines that a slope of the representative sample of the speech signal is generally flat.

33. The system according to claim 29 where the second filter increases a slope of the spectral response of the speech signal to approach a more sloped spectral response than the flat spectral response in preparation for prospective speech coding.

34. The system according to claim 29 where the evaluator comprises an averaging unit is adapted to average the accumulated samples over the minimum sampling duration to obtain the representative sample.

35. The system according to claim 29 where the evaluator assumes the spectral response of a speech signal is sloped in accordance with the defined characteristic slope prior to the expiration of the minimum sampling duration.

36. The system according to claim 29 where the defined characteristic slope approximately represents a Modified Intermediate Response System.

37. The system according to claim 29 where the evaluator triggers an adjustment of at least one encoding parameter to a revised encoding parameter

during the encoding process, the at least one encoding parameter affiliated with one of the first filter and the second filter.

38. The system according to claim 29 where the evaluator is coupled to an encoder, where the evaluator sends a flatness or slope indicator to the encoder for controlling coding parameters of a group consisting of pitch gains per frame or subframe, at least one filter coefficient of a perceptual weighting filter of the encoder, at least one filter coefficient of a synthesis filter of the encoder, at least one bandwidth expansion constant associated with a synthesis filter of at least one of the encoder and a decoder, at least one bandwidth expansion constant associated with a synthesis filter of a decoder, at least one bandwidth expansion constant associated with an analysis filter of an encoder, and at least one filtering coefficient associated with a post filter coupled to a decoder for performing an inverse signal processing operations with respect to the encoder.